International Journal on Recent and Innovation Trends in Computing and Communication ISSN: 2321-8169
Volume: 4 Issue: 7 127 - 133

A Modified Cross Correlation Double Talk Detector using Variable Threshold for Acoustic Echo Cancellation

Harkirat Kaur
Department of ECE
Punjabi University, Patiala
Punjab, India
kaurharkirat0@gmail.com

Rupinder Kaur
Department of ECE
Punjabi University, Patiala
Punjab, India
reepu_89@yahoo.com

Pulkit Sharma
Department of ECE
Punjabi University, Patiala
Punjab, India
sharmapulkit26@yahoo.com

Abstract — The presence of echo in the communication systems reduces the speech quality and can be overcome by using Acoustic Echo Cancellers (AEC). Acoustic Echo Canceller is a special device which estimates the echo and subtracts it from the microphone signal. Adaptive filters are used for this purpose. Adaptive filters diverge when in addition to far end signal, near end signal is also present. This situation is known as double talk which is handled by double talk detectors (DTD). Cross correlation based DTD is an attractive approach in which cross correlation between two signals give decision parameter. This parameter is compared to constant threshold to give decision. The constant threshold based double talk detector increases the probability of missed detection and increases the residual error of echo canceller. Thus degrades the performance of Acoustic echo canceller. In this paper, a new variable threshold based double talk detector is proposed in which a threshold value is evaluated with respect to the power of near end signal and far end signal. The proposed method shows the reduced probability of missed detections and mean square error, hence improving AEC performance. For updating of the filter coefficients, Variable Step Size Least Mean Square algorithm is used.

Keywords- *Acoustic Echo, Acoustic Echo canceller (AEC),Constant threshold based DTD,Variable threshold based DTD,MSE,Probability of Missed detection*

*****

I. INTRODUCTION

In communication systems, occurrence of echo is very common. When an object is placed at some distance from an audio source, the generated sound is reflected by that object. The reflected signal is the echo signal [12]. In communication technologies, two types of echoes are present i.e. Hybrid echo and acoustic echo. Hybrid echo is produced in wired telephonic systems due to impedance mismatching at the hybrids in the transmission lines. It is also called line echo or electrical echo [15]. Acoustic echo may occur whenever a loudspeaker and microphone are placed close to each other. It is one of the types of echo which occurs due to the poor coupling of voice between loudspeaker and microphone in hands free devices.

A person on the far end speaks in microphone and his/her voice is generated by loudspeaker on the near side. That sound wave after being reflected by nearby objects like walls, door, etc within the room is taken by microphone at near end. This result of reflections leads to creation of multipath echo which is transmitted back to the far end. Therefore the person who has spoken some words will hear his/her own voice in the form of echo. Thus, acoustic echo acts as a disturbance to users and degrades the quality of voice during conversation.

Cancellation of echo is required to provide echo free environment for improving the voice quality. The application of Acoustic Echo Cancellers is vital to provide the better quality of service [16].At each end of communication system AEC is connected. The basic principle behind AEC is the estimation of echo and subtraction of echo. The main purpose of basic echo canceller setup is that the near end signal is given to microphone and transmitted to far side when far end signal is produced by loudspeaker in near end room.

Adaptive filter is a digital filter which is used for the cancellation of echo. According to the optimization algorithm, this filter varies its parameters to minimize the error signal. Error signal is the difference between desired signal and adaptive filter output. The presence of near end signal along with far end signal results into the divergence of adaptive filter. This situation is called double talk. The near end signal acts as source of disturbance in adaptation of filter. During presence of near talk, error signal will consist of both estimation error and near end signal. If this signal is used for filter adaptation, then adaptive filter will move away from its optimal solution. Therefore, it is important to know whether near end signal is present or not and when the filter coefficient adaptation should stop. This work is done by Double Talk Detectors.

Figure-1 indicates the basic Acoustic Echo Canceller with a Double Talk Detector. The main aim of AEC is to estimate the echo and subtract it from received signal at microphone.

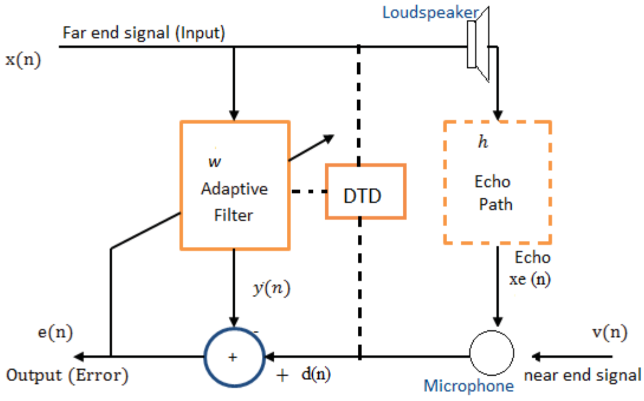

Figure-1 A Typical DTD based Acoustic Echo Canceller

Far end signal, x(n) is an input to AEC. This far end signal is passed through adaptive filter and echo path. After passing through echo path, echo signal, xe(n) is generated which is added up with a near end signal, v(n). This summation of signal is given to microphone and is treated as received signal. Adaptive filter generates estimated echo as its output which is subtracted from the received signal. This difference gives estimated error which is fed back to adaptive algorithm for the adaptation of weights. Ideally, this error must match with near end signal but due to non-linearity and residual echo, error signal is not exactly same as near end signal. This error signal is the output of acoustic echo canceller (AEC).

Double Talk Detectors (DTD) aim to estimate whether double talk exists and it controls the adaptation of filter weights to prevent the filter from diverging.

The performance of Acoustic Echo canceller depends on the algorithms we have used for filter adaptation and for detection double talk. In the previous research work, various methods for double talk detection have been proposed. Geigel method [14] provides energy level comparison of far end signal and received signal. This method is not much efficient and has less accuracy. A variance impulse response method [5] is another approach for DTD in which variance of maximum value of recent tap is estimated as a decision parameter. This method is lesser preferable because of high complexity and cost. These methods use a constant value of threshold which is compared with a decision statistics. Previous work does not focus on estimating the rate of missed detections and false alarms [9].

Various adaptive algorithms have been presented in [13] each of adaptive algorithms affects the AEC output. LMS algorithm and VSLMS algorithm have been mentioned in [9] and shows that VSLMS is better because it gives higher value of ERLE (Echo Return Loss Enhancement). Higher is the ERLE, better will be AEC.

After the introduction, the remaining of this paper is organized as follows: Section II presents the cross correlation method of DTD using a constant threshold and a new modified version using a variable threshold which is a proposed method is presented in section III. Section IV gives summary of Adaptive algorithm we used for implementation. Section V and VI presents various performance measures and simulation results obtained during implementation. Section VII concludes the paper.

II. CONSTANT THRESHOLD BASED DTD

In this paper, we have used cross correlation based double talk detector. Cross correlation between received signal, d(n) and far end signal, x(n) is used instead of cross correlation between far end signal and error signal. The received signal is given as:

$$d(n) = h^T x(n) + v(n) \tag{1}$$

Multiplying both sides of equation 4.1with far end signal, x(n) and taking expectation, equation becomes

$$E[d(n)x^T(n)] = h^T E[x(n)x^T(n)] + E\,[v(n)x^T(n)] \tag{2}$$

As far end signal, x(n) and near end signal, v(n) are uncorrelated. Therefore

$$E\,[v(n)x^T(n)] = 0 \tag{3}$$

$$E[d(n)x^T(n)] = h^T E[x(n)x^T(n)] \tag{4}$$

Again Multiplying equation (1) with received signal, d(n) and taking expectation, we get

$$E[d(n)d^T(n)] = h^T E[x(n)d^T(n)] + E[v(n)d^T(n)] \tag{5}$$

When near end signal is absent i.e. v(n)=0, equation(5) becomes

$$E[d(n)d^T(n)] = h^T E[x(n)d^T(n)] \tag{6}$$

Multiply equation (4) and eq. (6), we get

$$\{E[d(n)x^T(n)]\}^2 = E[x(n)x^T(n)]\, E[d(n)d^T(n)] \tag{7}$$

$$R_{dx}{}^2 = E[d^2(n)]\, E[x^2(n)] \tag{8}$$

Where, R_{dx} is the cross correlation between far end signal and received signal and is given as

$$R_{dx} = E[\, x(n)d^T(n)] \tag{9}$$

The cross correlation coefficient vector between x and d is given as:

$$\rho_{dx} = \frac{E\{x(n)d(n)\}}{\sqrt{E\{x^2(n)\}E\{d^2(n)\}}} \tag{10}$$

$$= \frac{R_{dx}}{\sigma_x \sigma_d}$$

Decision parameter is $p(n) = \left\|\rho_{dx}\right\| = \max\left|\rho_{dx}^i\right|$, (11)

$$i = 0,1 \ldots L-1$$

In above mentioned method of cross correlation, if p(n) is smaller than threshold, double talk is decided by detector. This method uses only constant or fixed threshold. The problem with constant threshold is that large number of detections is missed and some are taken as false alarms. For a fixed threshold method, we have to check various constant threshold values and to choose one value which gave best results. So to improve the detector performance, a variable threshold based method is employed. In this method, Threshold value is varied with respect to signal power and it is compared to the decision parameter.

III. VARIABLE THRESHOLD BASED DTD

This is the proposed method of double talk detection in this research work. A formula is derived for the estimation of variable threshold.

The cross correlation between far end signal and Received signal is given in terms of power but power calculation requires forgetting factor for smoothing which is difficult to analyze. Therefore, we use expectation instead of forgetting factor involvement.

$$\rho_{dx} = \frac{P_{dx}(n)}{\sqrt{P_d(n)P_x(n)}} = \frac{E\{x(n)d(n)\}}{\sqrt{E\{x^2(n)\}E\{d^2(n)\}}} \tag{12}$$

Firstly $P_{dx}(n)$,

ISSN: 2321-8169
127 - 133

$$P_{dx}(n) = E[d(n) * x(n)]$$

$$= E[(xe(n) + v(n)) * x(n)]$$

$$= E[xe(n).x(n)] + E[v(n).x(n)]$$

As x(n) and v(n) are uncorrelated, equation becomes

$$P_{dx}(n) = E[xe(n).x(n)]$$

$$P_{dx}(n) = h.E[x^T(n).x(n)]$$

$$P_{dx}(n) = h.\ \sigma_x^2 \quad (13)$$

$P_d(n)$, $\quad P_d(n) = E[d(n).d(n)]$

$$= E[(xe(n) + v(n)).(xe(n) + v(n))]$$

$$\approx E[h^T x(n).h\, x^T(n) + v^2(n)]$$

$$= h.h^T E[x(n)x^T(n)] + E[v^2(n)]$$

$$P_d(n) = h.h^T \sigma_x^2 + \sigma_v^2 \quad (14)$$

$P_x(n)$, $\quad P_x(n) = E[x(n).x^T(n)] = \sigma_x^2 \quad (15)$

Finally, combine eq. (13), (14), (15) and put in equation (12)

$$\rho_{dx} = \frac{P_{dx}(n)}{\sqrt{P_d(n)P_x(n)}} = \frac{h.h^T\ \sigma_x^2}{\sqrt{(h.h^T\sigma_x^2 + \sigma_v^2).\sigma_x^2}}$$

$$= \frac{1}{\sqrt{1 + \frac{\sigma_v^2}{h.h^T\sigma_x^2}}} \quad (16)$$

Equation (16) gives theoretical cross correlation which is exact value. This equation is modified for simplicity and is given as:

$$\rho_{dx} = \frac{1}{\sqrt{1 + \frac{\sigma_d^2 - \sigma_{xe}^2}{\sigma_x^2 h.h^T}}} \quad (17)$$

By approximating, value of h.h' is taken as 1. So, a variable threshold in double talk detector is calculated as follows:

$$T(i) = \frac{c}{\sqrt{1 + \frac{\sigma_d^2 - \sigma_{xe}^2}{\sigma_x^2}}} \approx \frac{c}{\sqrt{1 + \frac{\sigma_v^2}{\sigma_x^2}}} \quad (18)$$

c is some constant value which is taken as 1000 in simulations. From equation (16), the variable threshold is clearly a function of the ratio of near end power and far end power. In this paper, we have used this variable threshold based Double Talk Detector and comparing it with constant threshold based Double Talk Detector. With this new variable threshold, we can faster determine that double talk is over and detector easily decides whether double talk has occurred. Threshold is estimated for each iteration. Decision parameter is compared to this value for deciding double talk. A flowchart for Cross correlation based Double talk Detector which shows the various steps for simulation of detector is mentioned in figure-2.

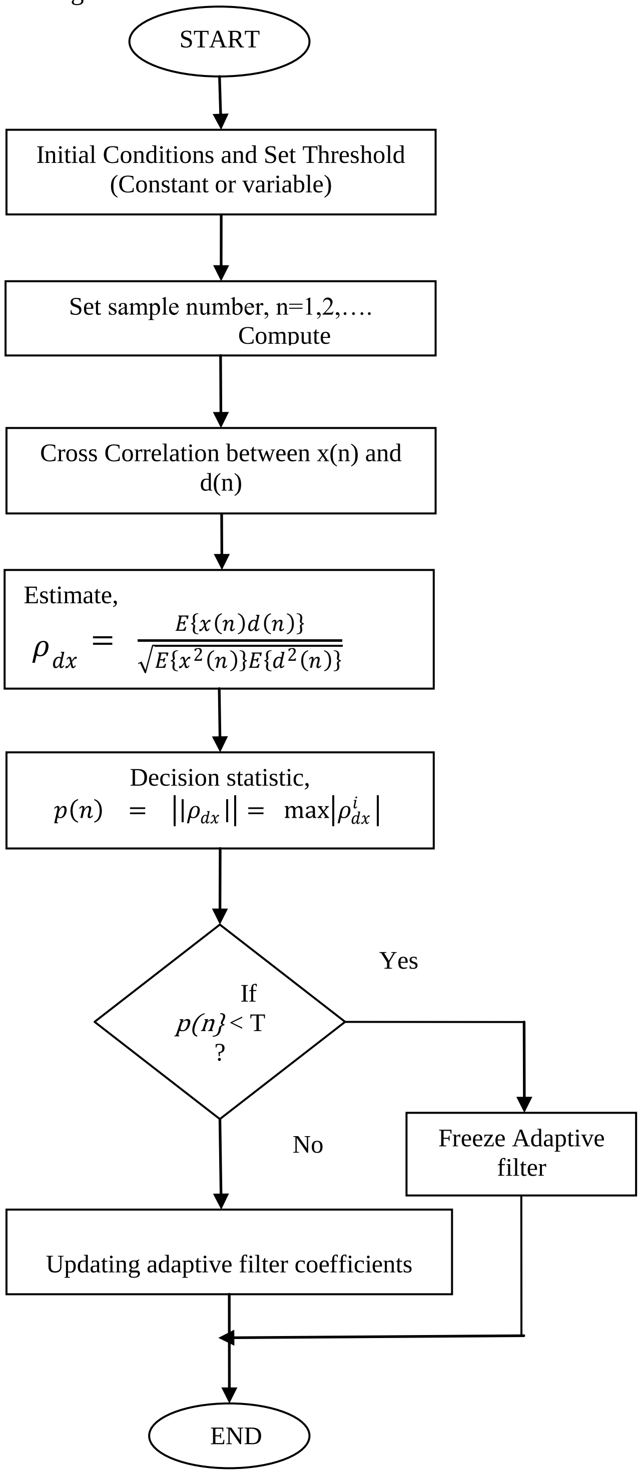

Figure-2 Flowchart for cross correlation based Double Talk Detector

IV. FILTER ADAPTATION

Double Talk detector is used in the acoustic echo canceller in which tap weight adaptation is done with the help of

VSLMS algorithm. In the general LMS algorithm, step size is constant value. The problem with this algorithm is that if step size is small, adaptive filter takes long time to converge to the optimal solution. On the other hand, if step size chosen is large, then adaptive filter diverges and system becomes unstable. To overcome this problem, VSLMS is used. Summary for VSLMS adaptive algorithm is shown in Table-1.

TABLE-1 SUMMARY FOR VSLMS ALGORITHM

Variable Step Least Mean Square (VSLMS) Algorithm	
Initial Conditions:	Length of adaptive filter: L Input vector: $x_{L,1} = [0,0 \ldots..0]^T$ Weight vector: $w_{L,1} = [0,0 \ldots..0]^T$ $0 < \alpha < 1$ and $\gamma > 0$
For each instant of time, n=1,2,…, Compute:	
Output signal :	$y(n) = w^T(n)x(n)$
Estimation error:	$e(n) = d(n) - y(n)$
Step size :	$\mu(n+1) = \alpha\mu(n) + \gamma e^2(n)$
Tap weight Adaptation:	$w(n+1) = w(n) + 2\mu(n)e(n)x(n)$

V. PERFORMANCE EVALUATION

The performance of AEC and DTD is evaluated in terms of some measures which are explained below:

A. *Echo Return Loss Enhancement (ERLE)*

ERLE is a measure of amount of echo cancelled by an echo canceller. It measures the loss that is introduced by only adaptive filter and depends on the design of algorithm. Mathematically, it is described as the ratio of power of microphone signal and power of residual error signal, immediately after cancellation [9]. It is expressed in decibels.

$$ERLE(dB) = 10log_{10}\frac{Power\ of\ Received\ signal}{Power\ of\ error\ signal}$$

$$= 10log_{10}\frac{P_d(n)}{P_e(n)}$$

Calculation of ERLE is performed in the absence of near end signal, in the portion when only echo is present in received signal [7], [13].Higher is the value of ERLE, better is performance of AEC.

B. *Mean Square Error (MSE)*

The algorithm used for echo cancellation should be such that which minimizes the mean square error [16].MSE provides the expected value of square of estimated error in acoustic echo canceller. To evaluate the performance, the graph and value of this parameter is important. It is given as

$$MSE = E\{e^2(n)\}$$

C. *Signal to Noise ratio (SNR)*

Signal to noise ratio is defined as the ratio of power of signal to the power of noise and is expressed in decibels (dB). It provides the power level comparison of a desired signal and background noise present in the signal.

D. *Near to Far end Ratio (NFR)*

It is defined as the ratio of near end speech power and power of far end speech. It is expressed in decibels.

$$NFR(dB) = 10log_{10}\frac{Power\ of\ near\ end\ signal}{Power\ of\ far\ end\ signal}$$

$$= 10log_{10}\frac{P_v(n)}{P_x(n)}$$

E. *DTD Performance*

The performance of double talk detectors can be evaluated using different measures. DTD can be viewed as a binary detection scheme. The basic characteristics of this scheme are:

Probability of detection (Pd): It is the probability of successful detection when signal or target (double talk) is present. Or it indicates the probability of detecting double talks when they actually exist.

Probability of missed detection (Pm= (1-Pd)): It is the probability when some detection is missed by the detector or probability of detection failure when double talk is present. The formula which is used in this research work is mentioned below:

$$Pm = 1 - \frac{\sum(FED).(NED).(DTD)}{\sum(FED).(NED)} = 1 - Pd$$

Where, FED is the output of far end speech detector, NED is the output of near end speech detector and DTD is the output of double talk detector.

Probability of false alarm (Pf): It is the probability of declaring detection when double talk is not present. Mathematical expression for probability of false alarm is given as:

$$Pf = \frac{\sum(FED)\,(DTD)}{length\ of\ far\ end\ speech}$$

These probabilities are used to evaluate the performance of double talk detector. For a well performed detector missed probabilities and false alarms must be mini-mum. It should have capability of detecting all possible double talks. If these factors improve the Double talk detector, ultimately it affects the performance of echo canceller.

VI. SIMULATION AND RESULTS

For simulation purpose, MATLAB R2013 platform is used. The results are obtained for 88000 samples and with an adaptive filter length of 1000. Input signals i.e. far end signal and near end signal of 11 seconds each in the form of wav files are loaded into MATLAB at the sampling rate of 8000Hz.The

 ISSN: 2321-8169
127 - 133

graphs plotted have x-axis denoting the number of samples and y-axis denoting the amplitude of the signal.

MATLAB Simulation results are shown in following figures.Figure-3 indicates input far end signal and figure-4 indicates near end signal. White Gaussian noise is added to the far end signal. Echo signal is added up with the near end signal to generate received signal shown in figure-5.

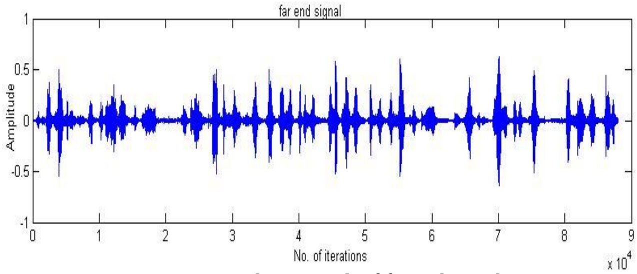

Figure-3 Simulation result of far end signal

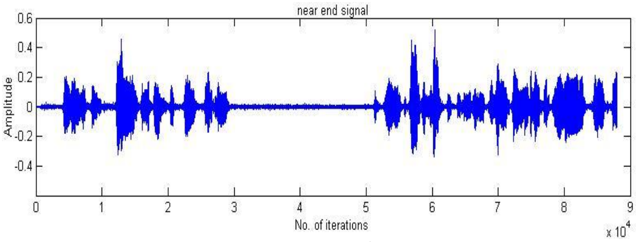

Figure-4 simulation result of near end signal

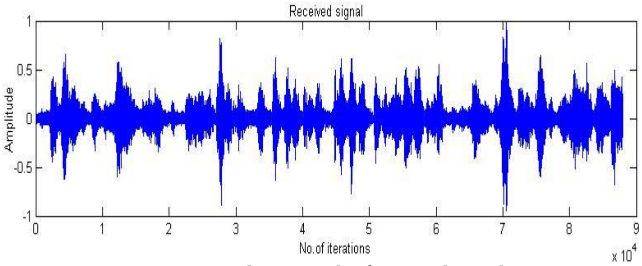

Figure-5 simulation result of received signal

A. *Results for Constant Threshold based DTD*

Double Talk Detector results into binary output which is given as ‘1’ if the estimated decision parameter is less than constant threshold indicating double talk is present. On the other hand detector output is ‘0’ if decision parameter is greater than constant threshold. Simulation result for DTD output as generated by using a constant threshold value of 490 is shown in figure - 6.

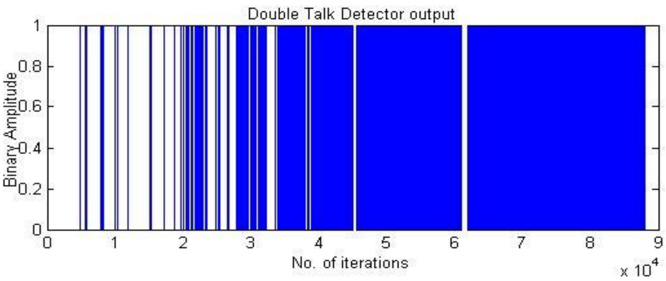

Figure-6 Double Talk Detection

Simulation result of Mean Square Error for constant threshold based DTD is mentioned in figure-7. MSE must approach minimum value.

Probability of missed detection is estimated by varying the near end signal energy and resultant plot is given in figure-8. We have used six different near end speech and NFR is calculated corresponding to them.

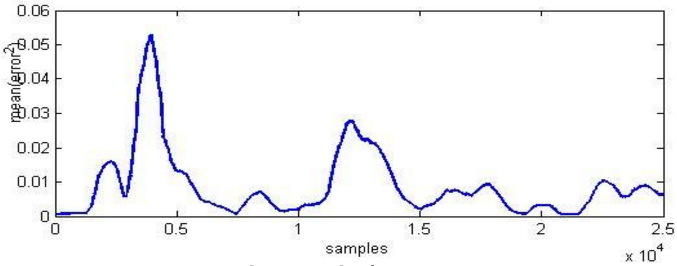

Figure-7 Simulation result of Mean Square Error

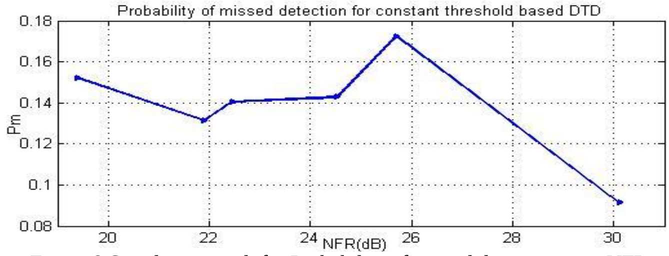

Figure-8 Simulation result for Probability of missed detection w.r.t NFR

B. *Results for Variable Threshold based DTD*

In the proposed method of Double Talk detection, threshold is estimated for each iteration and is compared to the decision parameter. The constant threshold based DTD gives results and shows that it gives poor Double talk detection. As many detections has been missed and large value of MSE. The simulation result for double talk detector output is shown in figure-9 and shows that it takes correct decisions.

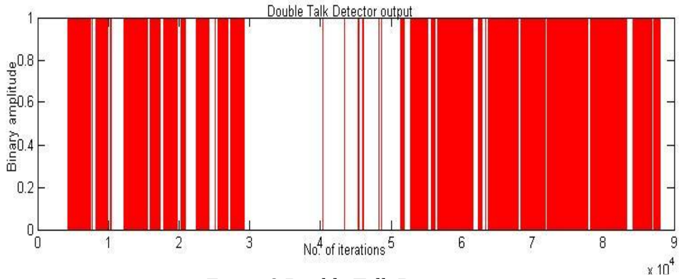

Figure-9 Double Talk Detection

As compared to Constant Threshold DTD the proposed method gives smaller values of MSE which is shown in figure-10. Maximum peak in graph is at 0.034 but in previous method this peak is at 0.05.

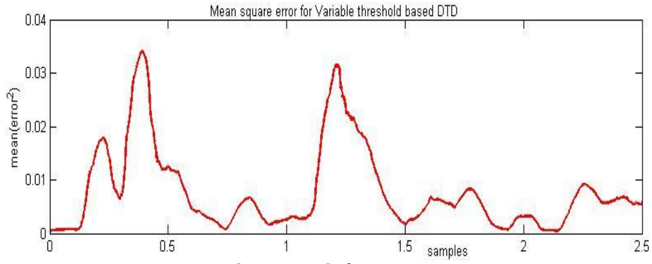

Figure-10 Simulation result for Mean Square Error

The variable threshold based DTD has smaller probability of missed detection. Simulation result for Probability of missed detection w.r.t NFR is shown in figure-11

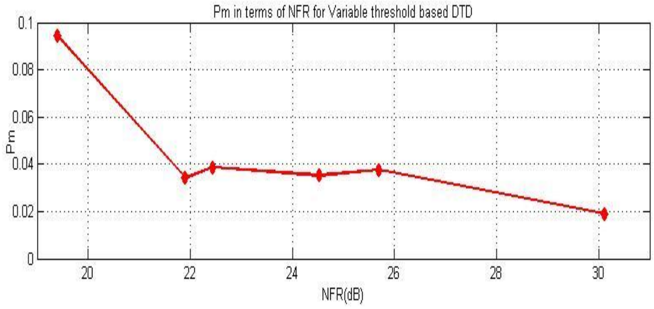

Figure-11 Simulation result for Probability of missed Detection w.r.t NFR

The optimum values of parameters used in the simulation of DTD based AEC are mentioned in the table-2.

Far_th and near_th are the threshold values of far end speech detector and near end speech detector respectively which are used for the estimation of probability of detections and missed detections. NFR is the near to far end speech ratio and its range is 19 to 3 db. L is the length of adaptive filter and N is the number of iterations or signal length. The estimated variable threshold has maximum value as 1000. This implementation is done with signal to noise ratio of 40 decibels.

TABLE-2 OPTIMAL VALUES OF PARAMETERS USED IN SIMULATION

Parameter	**Value**	**Parameter**	**Value**
N	88000	T(constant threshold)	490
L	1000	SNR(dB)	40
Fs	8000	Near_th	0.0001
γ	0.025	Far_th	0.0001
σ	0.97	NFR (dB)	19.39

Both methods of Double Talk Detectors are compared on the basis of parametric values which are shown in the table-3. It is clearly mentioned in the table that performance of proposed method results into better echo canceller.

The performance of AEC using variable threshold DTD is improved in terms of more ERLE, reduced rate of missed detections and lesser Mean Square error value. If probability of detection is increased, there is increase in false alarms. For constant threshold based DTD, probability of false alarm is 0.23 which increases to 0.26 for variable threshold based DTD. There is also increase of probability of successful detections. There is tradeoff between probability of detection and false alarms.

TABLE-3 COMPARISON OF RESULTS

	Constant Threshold based DTD (T=490)	**Variable Threshold based DTD**
Max ERLE (dB))	48	58
MSE	0.01	0.008
Probability of Detection (Pd)	0.84	0.88
Probability of missed Detection(Pm)	0.15	0.10
Probability of False Alarm (Pf)	0.23	0.26

VII. CONCLUSION

In this paper, cross correlation Double talk detector is modified by using a variable threshold instead of using constant threshold. The variable threshold depends on the power of far end signal and power of near end signal. Both methods of Double Talk Detector are compared and we concluded that variable threshold based DTD is better than constant threshold based DTD. The performance of AEC using variable threshold DTD is improved in terms of more ERLE, reduced rate of missed detections and lesser Mean Square error value.

REFERENCES

[1] Akihiko sugiyama, Jerome berclaz, miki sato, “Noise-Robust Double-Talk Detection Based On Normalized Cross Correlation And A Noise Offset” , International Conference on Acoustics, Speech, and Signal Processing - ICASSP , pp. iii/153-iii/156 Vol., 2005

[2] Per Åhgren, “Acoustic Echo Cancellation and Doubletalk Detection Using Estimated Loudspeaker Impulse Responses”, IEEE transactions on speech and audio processing, vol. 13, no. 6 November, 2005.

[3] Akihiko Sugiyama, J´Erome Berclaz†, Miki SatoJ. Benesty, T. Gänsler, “The Fast Cross-Correlation Double-Talk Detector”. Signal Processing, vol. 86, No. 6, pp. 1124-1139, Elsevier, 2006

[4] Wonchul Heo, Taehwan Kim, And Keunsung Bae, “Robust Double-Talk Detection In The Acoustic Echo Canceller Using Normalized Error Signal Power”, International conference on acoustics, speech and signal processing,IEEE,2007

[5] Sonika and Sanjeev Dhull, “Double Talk Detection in Acoustic Echo Cancellation based on Variance Impulse Response”, International Journal of Electronics and Communication Engineering, vol. 4, No. 5 , pp. 537-542, 2011

[6] Ivan J.Tashev, “Coherence based double talk detector with soft decision”, International conference on acoustics, speech and signal processing, IEEE, 2012

[7] V. K. Gupta, Mahesh Chandra and S. N. Sharan, "Acoustic Echo and Noise Cancellation System for Hand-Free Telecommunication using Variable Step Size Algorithms", Radio engineering, vol. 22, no. 1, pp.200-207, 2013.
[8] Mahfoud Hmidia, Abderrahmane amrouche, "Double Talk Detector based on speech feature extraction for acoustic echo cancellation", International Conference on software, telecommunication and computer networks, IEEE, 2014.
[9] Vineeta Das, Asutosh Kar ,Mahesh Chandra , " A new cross correlation based double talk detection algorithm for Nonlinear Acoustic Echo Cancellation ," TENCON 2014- IEEE region 10 Conference,2014.
[10] Mahfoud Hmidia, Abderrahmane amrouche, "A new structure for acoustic echo cancellation in double talk scenario using auxiliary filter", 14th International workshop on acoustic signal enhancement, IEEE, 2014.
[11] Ein Gyin Pwint, Su Su Yi Mon, Hla Myo Tun, "Design And Simulation Of An Acoustic Echo Cancellation System For Hand-Free Telecommunication", International Journal Of Scientific & Technology Research Volume 4, Issue 06, June 2015.
[12] Abhishek Deb, Asutosh Kar, Mahesh Chandra, "A Technical Review on Adaptive Algorithms for Acoustic Echo Cancellation", International Conference on Communication and Signal Processing, IEEE April 3-5, 2014.
[13] Mahesh Chandra, Asutosh Kar, Pankaj Goel, "Performance Evaluation of Adaptive Algorithms for Monophonic Acoustic Echo Cancellation: A Technical Review", International Journal of Applied Engineering Research, vol. 9, no-17, pp. 3781-3806, 2014.
[14] Harsh Kumar and Sanjeev Dhull, "Performance Analysis of Different DTD Methods for AEC", International Journal for Scientific Research & Development| Vol. 3, Issue 06, 2015.
[15] Srinivasaprasath Raghavendran, "Implementation of an acoustic echo canceller using MATLAB", Master's thesis, University of South Florida, 2003.
[16] Hung Ngoc Nguyen, Majid Dowlatnia, Azhar Sarfraz, "Implementation of the LMS and NLMS algorithms for Acoustic Echo Cancellation in teleconference system", Master's thesis, vaxjo university, 2009.
[17] J.G. Proakis and D.G. Manolakis, Digital Signal processing Principles, Algorithms and Applications 4th edition prentice-hall,2006.
[18] Harkirat Kaur, Rupinder Kaur and Pulkit Sharma , " A survey on Double Talk Detection based Acoustic Echo Cancellation" , International Conference On Sciences, Engineering And Technical Innovations,2016.
[19] Thamer M. Jamel, "Performance Enhancement Of Acoustic Echo Canceller Using New Time Varying Step Size LMS Algorithm(NVSSLMS)" ,International Journal of advancements in computing technology, Korea, Vol.3, NO. 1, January 2013.
[20] E.Hari Krishna, M.Raghuram, K. Venu Madhav and K. Ashoka Reddy, "Acoustic echo cancellation using a computationally efficient Transform domain LMS Adaptive Filter," 10th International Conference on information science, signal processing and their applications, *IEEE, 2010.*